

**Listing of the Claims:**

The following is a complete listing of all the claims in the application, with an indication of the status of each:

- 1        1 (Previously Amended). A speech coding apparatus including at least:  
2              a spectrum parameter calculation section for receiving a speech  
3              signal, obtaining a spectrum parameter, and quantizing the spectrum  
4              parameter,  
5              an adaptive codebook section for obtaining a delay and a gain from  
6              a past quantized sound source signal by using an adaptive codebook, and  
7              obtaining a residue by predicting a speech signal, and  
8              a sound source quantization section for quantizing a sound source  
9              signal of the speech signal by using the spectrum parameter and outputting  
10             the sound source signal, comprising:  
11              a discrimination section for discriminating a voiced sound mode  
12              and an unvoiced sound mode on a basis of a past quantized gain of an  
13              adaptive codebook;  
14              a sound source quantization section which has a codebook for  
15              representing a sound source signal by a combination of a plurality of non-  
16              zero pulses and collectively quantizing amplitudes or polarities of the  
17              pulses based on an output from said discrimination section, and searches  
18              combinations of code vectors stored in said codebook and a plurality of  
19              shift amounts used to shift positions of the pulses so as to output a  
20              combination of a code vector and shift amount which minimizes distortion  
21              relative to input speech; and  
22              a multiplexer section for outputting a combination of an output  
23              from said spectrum parameter calculation section, an output from said  
24              adaptive codebook section, and an output from said sound source  
25              quantization section.

1        2 (Previously Amended). A speech coding apparatus including at least:  
2              a spectrum parameter calculation section for receiving a speech  
3              signal, obtaining a spectrum parameter,  
4              an adaptive codebook section for obtaining a delay and a gain from  
5              a past quantized sound source signal by using an adaptive codebook, and  
6              obtaining a residue by predicting a speech signal, and  
7              a sound source quantization section for quantizing a sound source  
8              signal of the speech signal by using the spectrum parameter and outputting  
9              the sound source signal, comprising:  
10              a discrimination section for discriminating a voice soundmode and  
11              an unvoiced sound mode on a basis of a past quantized gain of an adaptive  
12              codebook;  
13              a sound source quantization section which has a codebook for  
14              representing a sound source signal by a combination of a plurality of non-  
15              zero pulses and collectively quantizing amplitudes or polarities of the  
16              pulses based on an output from said discrimination section, and outputs a  
17              code vector that minimizes distortion relative to input speech by generating  
18              positions of the pulses according to a predetermined rule; and  
19              a multiplexer section for outputting a combination of an output  
20              from said spectrum parameter calculation section, an output from said  
21              adaptive codebook section, and an output from said sound source  
22              quantization section.

1        3 (Previously Amended). A speech coding apparatus including at least:  
2              a spectrum parameter calculation section for receiving a speech  
3              signal, obtaining a spectrum parameter, and quantizing the spectrum  
4              parameter,  
5              an adaptive codebook section for obtaining a delay and a gain from a  
6              past quantized sound source signal by using an adaptive codebook, and  
7              obtaining a residue by predicting a speech signal, and

8            a sound source quantization section for quantizing a sound source  
9        signal of the speech signal by using the spectrum parameter and outputting  
10      the sound source signal, comprising:

11            a discrimination section for discriminating a voice sound mode and  
12        an unvoiced sound mode on the basis of a past quantized gain of an  
13      adaptive codebook;

14            a sound source quantization section which has a codebook for  
15        representing a sound source signal by a combination of a plurality of non-  
16       zero pulses and collectively quantizing amplitudes or polarities of the  
17       pulses based an output from said discrimination section, and a gain  
18       codebook for quantizing gains, and searches combinations of code vectors  
19       stored in said codebook, a plurality of shift amounts used to shift positions  
20       of the pulses, and gain code vectors stored in said gain codebook so as to  
21       output a combination of a code vector, shift amount, and gain code vector  
22       which minimizes distortion relative to input speech; and

23            a multiplexer section for outputting a combination of an output  
24        from said spectrum parameter calculation section, an output from said  
25       adaptive codebook section, and an output from said sound source  
26       quantization section.

1        4 (Previously Amended). A speech coding apparatus including at least:

2            a spectrum parameter calculation section for receiving a speech  
3        signal, obtaining a spectrum parameter, and quantizing the spectrum  
4       parameter,

5            an adaptive codebook section for obtaining a delay an a gain from a  
6        past quantized sound source signal by using an adaptive codebook, and  
7       obtaining a residue by predicting a speech signal, and

8            a sound source quantization section for quantizing a sound source  
9        signal of the speech signal by using the spectrum parameter and outputting  
10      the sound source signal, comprising:

11            a discrimination section for discriminating a voice sound mode and  
12        an unvoiced sound mode on the basis of a past quantized gain of an  
13        adaptive codebook;

14            a sound source quantization section which has a codebook for  
15        representing a sound source signal by a combination of a plurality of non-  
16        zero pulses and collectively quantizing amplitudes or polarities of the  
17        pulses based on an output from said discrimination section indicates a  
18        predetermined mode, and a gain codebook for quantizing gains, and  
19        outputs a combination of a code vector and gain code vector which  
20        minimizes distortion relative to input speech by generating positions of the  
21        pulses according to a predetermined rule; and

22            a multiplexer section for outputting a combination of an output  
23        from said spectrum parameter calculation section, an output from said  
24        adaptive codebook section, and an output from said sound source  
25        quantization section.

5 (Canceled).

1        6 (Previously Amended). A speech coding/decoding apparatus comprising:  
2            a speech coding apparatus including:  
3            a spectrum parameter calculation section for receiving a speech  
4        signal, obtaining a spectrum parameter, and quantizing the spectrum  
5        parameter,  
6            an adaptive codebook section for obtaining a delay and a gain from  
7        a past quantized sound source signal by using an adaptive codebook, and  
8        obtaining a residue by predicting a speech signal,  
9            a sound source quantization section for quantizing a sound source  
10      signal of the speech signal by using the spectrum parameter and outputting  
11      the sound source signal,  
12            a discrimination section for discriminating a voice sound mode and

13       an unvoiced sound mode on the basis of a past quantized gain of a adaptive  
14       codebook, and  
15                a codebook for representing a sound source signal by a  
16       combination of a plurality of non-zero pulses and collectively quantizing  
17       amplitudes or polarities of the pulses when an output from said  
18       discrimination section indicates a predetermined mode,  
19                said sound source quantization section searching combinations of  
20       code vectors stored in said codebook and a plurality of shift amounts used  
21       to shift positions of the pulses so as to output a combination of a code  
22       vector and shift amount which minimizes distortion relative to input  
23       speech, and further including  
24                a multiplexer section for outputting a combination of an output  
25       from said spectrum parameter calculation section, an output from said  
26       adaptive codeboook section, and an output from said sound source  
27       quantization section; and  
28                a speech decoding apparatus including at least:  
29                a demultiplexer section for receiving and demultiplexing a  
30       spectrum parameter, a delay of an adaptive codebook, a quantized gain,  
31       and quantized sound source information,  
32                a mode discrimination section for discriminating a mode by using a  
33       past quantized gain in said adaptive codebook,  
34                a sound source signal reconstructing section for reconstructing a  
35       sound source signal by generating non-zero pulses from the quantized  
36       sound source information when an output from said discrimination  
37       indicates a predetermined mode, and  
38                a synthesis filter section which is constituted by spectrum  
39       parameters and reproduces a speech signal by filtering the sound source  
40       signal.

1       7 (Previously Amended). A speech coding/decoding apparatus comprising:

2           a speech coding apparatus including:  
3            a spectrum parameter calculation section for receiving a speech  
4       signal, obtaining a spectrum parameter, and quantizing the spectrum  
5       parameter,  
6            an adaptive codebook section for obtaining a delay and a gain from  
7       a past quantized sound source signal by using an adaptive codebook, and  
8       obtaining a residue by predicting a speech signal,  
9            a sound source quantization section for quantizing a sound source  
10      signal of the speech signal by using the spectrum parameter and outputting  
11      the sound source signal,  
12           a discrimination section for discriminating a voice sound mode and  
13       an unvoiced sound mode on the basis of a past quantized gain of an  
14       adaptive codebook, and  
15           a codebook for representing a sound source signal by a  
16       combination of a plurality of non-zero pulses and collectively quantizing  
17       amplitudes or polarities of the pulses based on an output from said  
18       discrimination section,  
19           said sound source quantization section outputting a combination of  
20       a code vector and shift amount which minimizes distortion relative to input  
21       speech by generating positions of the pulses according to a predetermined  
22       rule, and further including  
23           a multiplexer section for outputting a combination of an output  
24       from said spectrum parameter calculation section, an output from said  
25       adaptive codebook section, and an output from said sound source  
26       quantization section; and  
27           a speech decoding apparatus including at least:  
28            a demultiplexer section for receiving and demultiplexing a  
29       spectrum parameter, a delay of an adaptive codebook, a quantized gain,  
30       and quantized sound source information,  
31           a mode discrimination section for discriminating a mode by using a

32        past quantized gain in said adaptive codebook,  
33            a sound source signal reconstructing section for reconstructing a  
34        sound source signal by generating positions of pulses according to a  
35        predetermined rule and generating amplitudes or polarities for the pulses  
36        from a code vector when an output from said discrimination section  
37        indicates a predetermined mode, and  
38            a synthesis filter section which includes spectrum parameters and  
39        reproduces a speech signal by filtering the sound source signal.

1        8 (Previously Amended). A speech coding apparatus comprising:  
2            a spectrum parameter calculation section for receiving a speech  
3        signal, obtaining a spectrum parameter, and quantizing the spectrum  
4        parameter;  
5            means for obtaining a delay and a gain from a past quantized sound  
6        source signal by using an adaptive codebook, and obtaining a residue by  
7        predicting a speech signal; and  
8            mode discrimination means for receiving a past quantized adaptive  
9        codebook gain and performing mode discrimination associated with a  
10      voiced/unvoiced mode by comparing the gain with a predetermined  
11      threshold, and  
12            further comprising:  
13            sound source quantization means for quantizing a sound source  
14        signal of the speech signal by using the spectrum parameter and outputting  
15        the signal, and searching combinations of code vectors stored in a  
16        codebook for collectively quantizing amplitudes or polarities of a plurality  
17        of pulses in a predetermined mode and a plurality of shift amounts used to  
18        temporally shift a predetermined pulse position so as to select a  
19        combination of an index of a code vector and a shift amount which  
20        minimizes distortion relative to input speech;  
21            gain quantization means for quantizing a gain by using a gain

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22 codebook; and

23 multiplex means for outputting a combination of outputs from said  
24 spectrum parameter calculation means, said adaptive codebook means, said  
25 sound source quantization means, and said gain quantization means.

1 9 (Original). An apparatus according to claim 8, wherein said sound source  
2 quantization means uses a position generated according to a predetermined  
3 rule as a pulse position when mode discrimination indicates a  
4 predetermined mode.

1 10 (Original). An apparatus according to claim 9, wherein when mode  
2 discrimination indicates a predetermined mode, a predetermined number of  
3 pulse positions are generated by random number generating means and  
4 output to said sound source quantization means.

1 11 (Original). An apparatus according to claim 8, wherein when mode  
2 discrimination indicates a predetermined mode, said sound source  
3 quantization means selects a plurality of combinations from combinations  
4 of all code vectors in said codebook and shift amounts for pulse positions  
5 in an order in which a predetermined distortion amount is minimized, and  
6 outputs the combinations to said gain quantization means, and

7               said gain quantization means quantized a plurality of sets of  
8 outputs from said sound source quantization means by using said gain  
9 codebook, and selects a combination of a shift amount, sound source code  
10 vector, and gain code vector which minimizes the predetermined distortion  
11 amount.